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(54) **SPATIAL FILTER BANK FOR HEARING SYSTEM**

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H04R 1/10 (2006.01)

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CPC *H04R 25/552* (2013.01); *H04R 25/407* (2013.01); *H04R 25/50* (2013.01); *H04R 1/1083* (2013.01); *H04R 25/43* (2013.01); *H04R 25/453* (2013.01); *H04R 25/554* (2013.01)

(58) **Field of Classification Search**
None
See application file for complete search history.

(56) **References Cited**

U.S. PATENT DOCUMENTS

6,987,856 B1	1/2006	Feng et al.	
8,526,647 B2 *	9/2013	Pedersen	H04R 25/407
			381/313
2003/0063759 A1 *	4/2003	Brennan	H04R 3/005
			381/92
2012/0093336 A1	4/2012	Said et al.	

FOREIGN PATENT DOCUMENTS

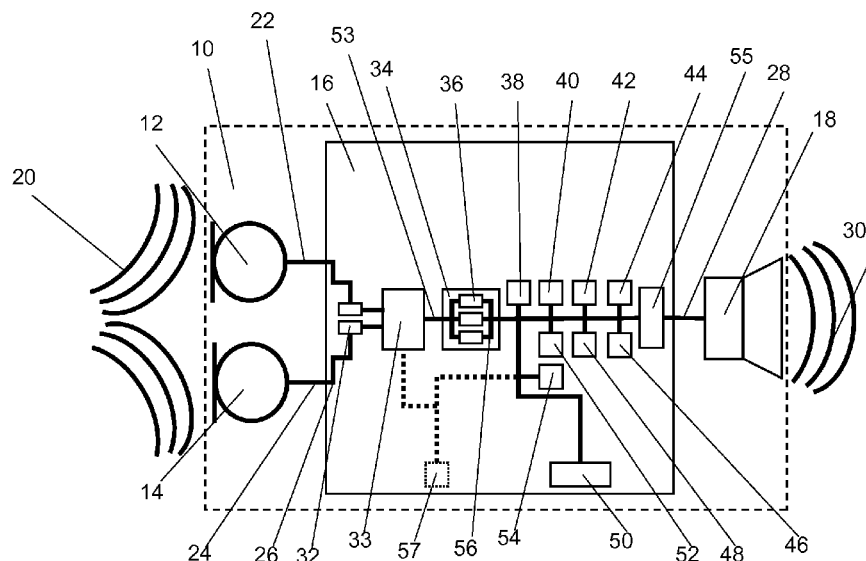|		
EP	1 962 547 A1	8/2008
WO	WO 03/015464 A2	2/2003

* cited by examiner

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(57) **ABSTRACT**

The present invention regards a hearing system configured to be worn by a user comprising an environment sound input unit, an output transducer, and electric circuitry. The environment sound input unit is configured to receive sound from the environment of the environment sound input unit and to generate sound signals representing sound of the environment. The output transducer is configured to stimulate hearing of a user. The electric circuitry comprises a spatial filterbank. The spatial filterbank is configured to use the sound signals to generate spatial sound signals dividing a total space of the environment sound in subspaces. Each spatial sound signal represents sound coming from a subspace. The subspaces may (in particular modes of operation) be either fixed, or dynamically determined, or a mixture thereof.
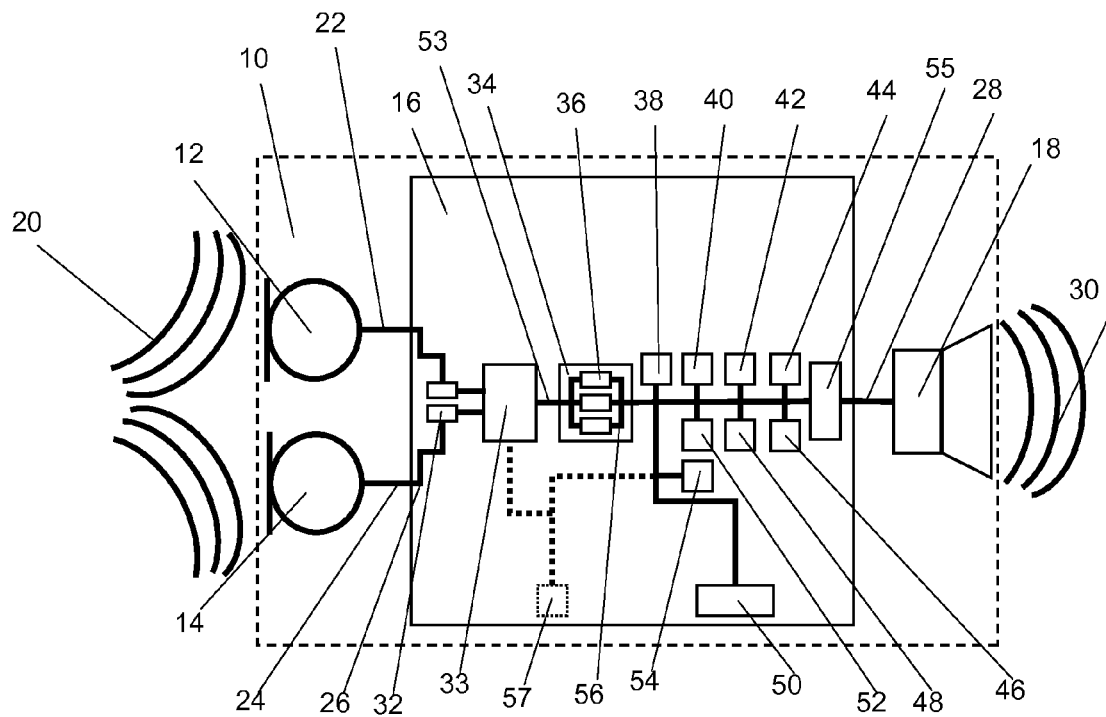
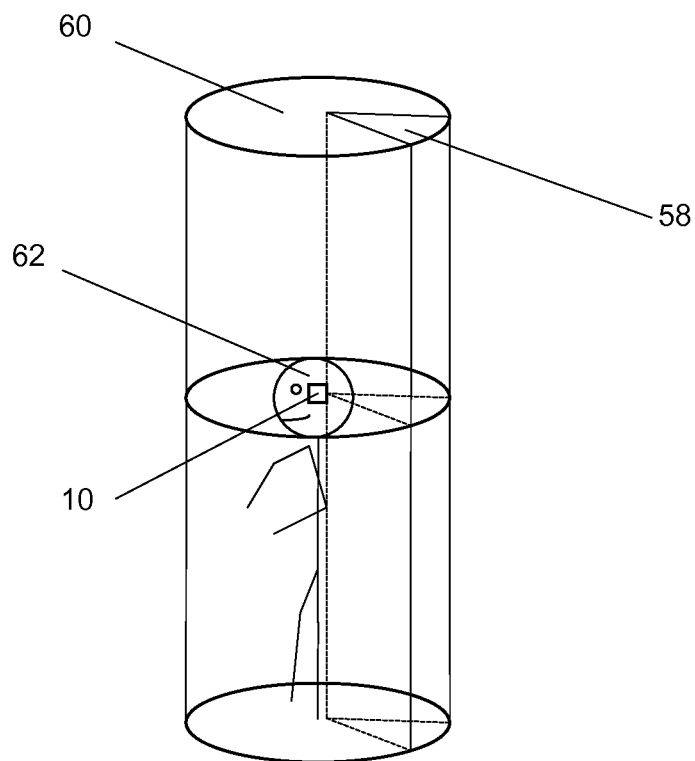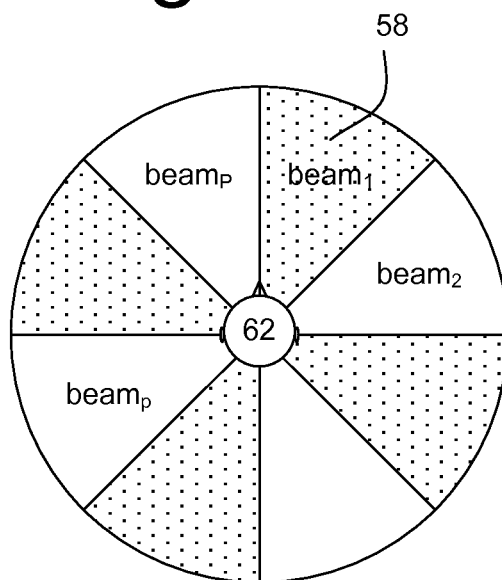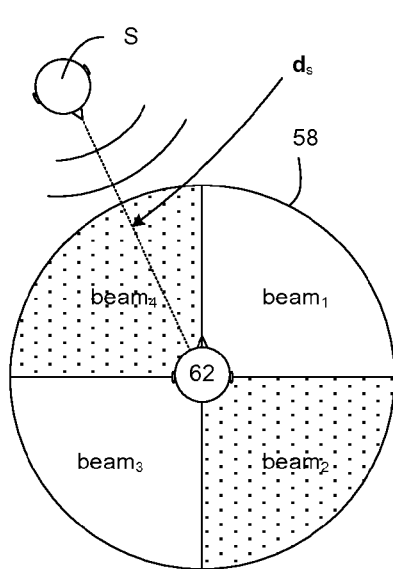**17 Claims, 4 Drawing Sheets**

Fig. 1

Fig. 2A

Fig. 2B

Fig. 2C

Fig. 2D

Fig. 2E

100

110

120

130

140
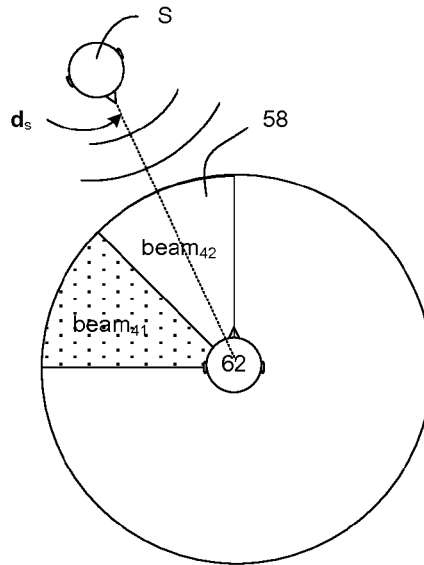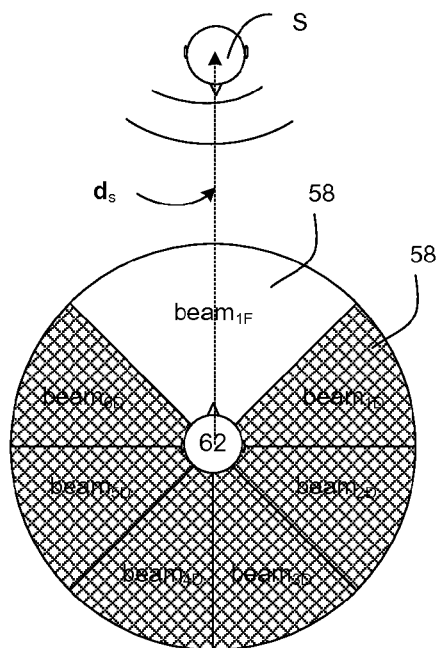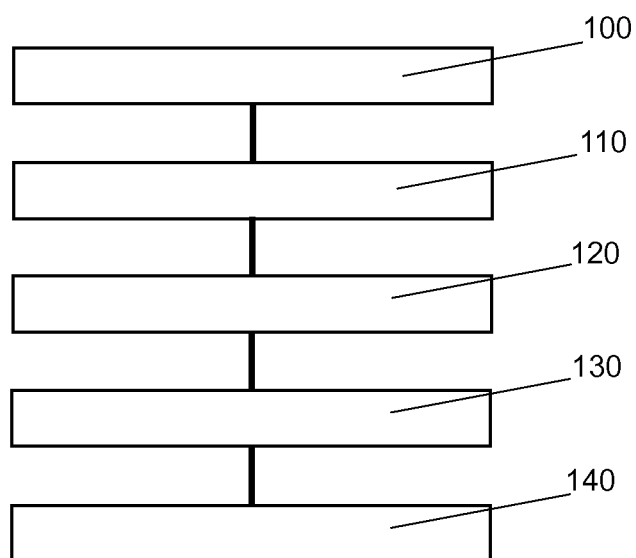
Fig. 3

SPATIAL FILTER BANK FOR HEARING SYSTEM

The invention regards a hearing system configured to be worn by a user comprising, an environment sound input unit, an output transducer, and electric circuitry, which comprises a spatial filterbank configured to divide sound signals in subspaces of a total space.

Hearing systems, e.g., hearing devices, binaural hearing aids, hearing aids or the like are used to stimulate the hearing of a user, e.g., by sound generated by a speaker or by bone conducted vibrations generated by a vibrator attached to the skull, or by electric stimuli propagated to electrodes of a cochlear implant. Hearing systems typically comprise a microphone, an output transducer, electric circuitry, and a power source. The microphone receives a sound and generates a sound signal. The sound signal is processed by the electric circuitry and a processed sound (or vibration or electric stimuli) is generated by the output transducer to stimulate the hearing of the user. In order to improve the hearing experience of a user, a spectral filterbank can be included in the electric circuitry, which, e.g., analyses different frequency bands or processes sound signals in different frequency bands individually and allows improving the signal-to-noise ratio. Spectral filterbanks are typically running online in many hearing aids today.

Typically, the microphones of the hearing system used to receive the incoming sound are omnidirectional, meaning that they do not differentiate between the directions of the sound. In order to improve the hearing of a user, a beamformer can be included in the electric circuitry. The beamformer improves the spatial hearing by suppressing sound from other directions than a direction defined by beamformer parameters. In this way the signal-to-noise ratio can be increased, as mainly sound from a sound source, e.g., in front of the user, is received. Typically, a beamformer divides the space in two subspaces, one from which sound is received and the rest, where sound is suppressed, which results in spatial hearing.

US 2003/0063759 A1 presents a directional signal processing system for beamforming information signals. The directional signal processing system includes a plurality of microphones, a synthesis filterbank, a signal processor, and an oversampled filterbank with an analysis filterbank. The analysis filterbank is configured to transform a plurality of information signals in time domain from the microphones into a plurality of channel signals in transform domain. The signal processor is configured to process the outputs of the analysis filter bank for beamforming the information signals. The synthesis filterbank is configured to transform the outputs of the signal processor to a single information signal in time domain.

U.S. Pat. No. 6,925,189 B1 shows a device that adaptively produces an output beam including a plurality of microphones and a processor. The microphones receive sound energy from an external environment and produce a plurality of microphone outputs. The processor produces a plurality of first order beams based on the microphone outputs and determines an amount of reverberation in the external environment, e.g., by comparison of the first order beams. The first order beams can have a sensitivity in a given direction different from the other channels. The processor further adaptively produces a second order output beam taking into consideration the determined amount of reverberation, e.g., by adaptively combining the plurality of first order beams or by adaptively combining the microphone outputs.

In EP 2 568 719 A1 a wearable sound amplification apparatus for the hearing impaired is presented. The wearable sound amplification apparatus comprises a first ear piece, a second ear piece, a first sound collector, a second sound collector, and a sound processing apparatus. Each of the first and second sound collectors is adapted for collecting sound ambient to a user and for outputting the collected ambient sound for processing by the sound processing apparatus. The sound processing apparatus comprises sound processing means for receiving and processing diversity sounds collected by the first and second sound collector using diversity techniques such as beamforming techniques. The sound processing apparatus further comprises means for subsequently outputting audio output to the user by or through one of or both the first and second ear pieces. The sound collectors are adapted to follow head movements of the user when the head of the user turns with respect to the body of the user.

It is an object of the invention to provide an improved hearing system.

This object is achieved by a hearing system configured to be worn by a user, which comprises an environment sound input unit, an output transducer, and electric circuitry. The environment sound input unit is configured to receive sound from the environment of the environment sound input unit and to generate sound signals representing sound of the environment. The output transducer is configured to stimulate hearing of a user. The electric circuitry comprises a spatial filterbank. The spatial filterbank is configured to use the sound signals to generate spatial sound signals dividing a total space of the environment sound in subspaces, defining a configuration of subspaces. Each spatial sound signal represents sound coming from a respective subspace. The environment sound input unit can for example comprise two microphones on a hearing device, a combination of one microphone on each of a hearing device in a binaural hearing system, a microphone array and/or any other sound input that is configured to receive sound from the environment and which is configured to generate sound signals from the sound which represent sound of the environment including spatial information of the sound. The spatial information can be derived from the sound signals by methods known in the art, e.g., determining cross correlation functions of the sound signals. Space here means the complete environment, i.e., surrounding of a user. A subspace is a part of the space and can for example be a volume, e.g. an angular slice of space surrounding the user (cf. e.g. FIGS. 2A-2E). The subspaces may but need not be of equal form and size, but can in principle be of any form and size (and location relative to the user). Likewise, the subspaces need not add up to fill the total space, but may be focused on continuous or discrete volumes of the total space around a user.

A specific 'configuration of subspaces' is in the present context taken to mean a specific 'geometrical arrangement of subspaces', as e.g. defined by one or more subspace parameters, which may include one or more of: a specific number of subspaces, a specific size (e.g. of a cross-sectional area or a volume) of the individual subspaces, a specific form (e.g. a spherical cone, or a cylindrical slice, etc.) of the individual subspaces, a location of the individual subspaces, a direction from the user (wearing the hearing system) to a point in space separated from the user defining an elongate volume (e.g. a cone). It is intended that a specific configuration of subspaces is defined by one of more subspace parameters as mentioned above or elsewhere in the present disclosure.

The spatial filterbank can also be configured to divide the sound signals in subspaces of the total space generating spatial sound signals. Alternatively, the electric circuitry can also be configured to generate a total space sound signal from the sound signals and the spatial filterbank can be configured to divide the total space sound signal in subspaces of the total space generating spatial sound signals.

One aspect of the invention is an improved voice signal detection and/or target signal detection, by performing a target signal detection and/or a voice activity detection on a respective spatial sound signal. Assuming that the target signal is present in a given subspace, the spatial sound signal of that subspace may have an improved target signal-to-noise signal ratio compared to sound signals which include the total space (i.e. the complete surrounding of a user), or other subspaces (not including the sound source in question). Further, the detection of several sound sources, e.g., talkers in different subspaces is possible by running voice activity detection in parallel in the different subspaces. Another aspect of the invention is that the location and/or direction of a sound source can be estimated. This allows to select subspaces and perform different processing steps on different subspaces, e.g., different processing of subspaces comprising mainly voice signals and subspaces comprising mainly noise signals. For example dedicated noise reduction systems can be applied to enhance the sound signals from the direction or directions of the sound source. Another aspect of the invention is that the hearing of a user can be stimulated by a spatial sound signal representing a certain subspace, e.g., a subspace behind the user, in front of the user, or at the side of a user, e.g., in a car-cabin situation. The spatial sound signal can be selected from the plurality of spatial sound signals, allowing to almost instantly switch from one subspace to another subspace, preventing the possible missing of the beginning of a sentence in a conversation, when the user first has to turn into the direction of the sound source or focus on the subspace of the sound source. A further aspect of the invention is an improved feedback howl detection. The invention allows an improved distinction between the following two situations: i) a feedback howl and ii) an external signal, e.g., a violin playing, which generates a similar sound signal as a feedback howl. The spatial filterbank allows to exploit the fact that feedback howls tend to occur from a particular subspace or direction, so that the spatial difference between a howl and the violin playing can be exploited for improved howl detection.

The hearing system is preferably a hearing aid configured to stimulate the hearing of a hearing impaired user. The hearing system can also be a binaural hearing system comprising two hearing aids, one for each of the ears of a user. In a preferred embodiment of a binaural hearing system, the sound signals of the respective environment sound inputs are wirelessly transmitted between the two hearing aids of the binaural hearing system. The spatial filterbank in this case can have a better resolution as more sound signals can be processed by the spatial filterbank, e.g., four sound signals from, e.g., two microphones in each hearing aid. In an alternative embodiment of a binaural hearing system detection decisions, e.g., voice signal detection and/or target signal detection, or their underlying statistics, e.g. signal-to-noise ratio (SNR) are transmitted between the hearing aids of the binaural hearing system. In this case the resolution of the respective hearing aid can be improved by using the sound signals of the respective hearing aid in dependence on the information received by the other hearing aid. Using the information of the other hearing aid instead of transmitting and receiving complete

sound signals decreases the computational demand in terms of bit rate and/or battery usage.

In a preferred embodiment the spatial filterbank comprises at least one beamformer. Preferably the spatial filterbank comprises several beamformers which can be operated in parallel to each other. Each beamformer is preferably configured to process the sound signals by generating a spatial sound signal, i.e., a beam, which represents sound coming from a respective subspace. A beam in this text is the combination of sound signals generated from, e.g., two or more microphones. A beam can be understood as the sound signal produced by a combination of two or more microphones into a single directional microphone. The combination of the microphones generates a directional response called a beampattern. A respective beampattern of a beamformer corresponds to a respective subspace. The subspaces are preferably cylinder sectors and can also be spheres, cylinders, pyramids, dodecahedra or other geometrical structures that allow to divide a space into subspaces. The subspaces preferably add up to the total space, meaning that the subspaces fill the total space completely and do not overlap, i.e., the beampatterns "add up to 1" such as it is preferably done in standard spectral perfect-reconstruction filterbanks. The addition of the respective subspaces to a summed subspace can also exceed the total space or occupy a smaller space than the total space, meaning that there can be empty spaces between subspaces and/or overlap of subspaces. The subspaces can be spaced differently. Preferably the subspaces are equally spaced.

In one embodiment the electric circuitry comprises a voice activity detection unit. The voice activity detection unit is preferably configured to determine whether a voice signal is present in a respective spatial sound signal. The voice detection unit preferably has at least two detection modes. In a binary mode the voice activity detection unit is configured to make a binary decision between "voice present" or "voice absent" in a spatial sound signal. In a continuous mode the voice activity detection unit is configured to estimate a probability for the voice signal to be present in the spatial sound signal, i.e., a number between 0 and 1. The voice activity detection unit can also be applied to one or more of the sound signals or the total space sound signal generated by the environment sound input. The detection whether a voice signal is present in a sound signal by the voice activity unit can be performed by a method known in the art, e.g., by using a means to detect whether harmonic structure and synchronous energy is present in the sound signal and/or spatial sound signal. The harmonic structure and synchronous energy indicates a voice signal, as vowels have unique characteristics consisting of a fundamental tone and a number of harmonics showing up synchronously in the frequencies above the fundamental tone. The voice activity detection unit can be configured to continuously detect whether a voice signal is present in a sound signal and/or spatial sound signal. The electric circuitry preferably comprises a sound parameter determination unit which is configured to determine a sound level and/or signal-to-noise ratio of a sound signal and/or spatial sound signal and/or if a sound level and/or signal-to-noise ratio of a sound signal and/or spatial sound signal is above a predetermined threshold. The voice activity detection unit can be configured only to be activated to detect whether a voice signal is present in a sound signal and/or spatial sound signal when the sound level and/or signal-to-noise ratio of a sound signal and/or spatial sound signal is above a predetermined threshold. The voice activity detection unit and/or the sound parameter

determination unit can be a unit in the electric circuitry or an algorithm performed in the electric circuitry.

In one embodiment the electric circuitry comprises a noise detection unit. The noise detection unit is preferably configured to determine whether a noise signal is present in a respective spatial sound signal. In an embodiment, the noise detection unit is adapted to estimate a level of noise at a given point in time (e.g. in individual frequency bands). The noise detection unit preferably has at least two detection modes. In the binary mode the noise detection unit is configured to make a binary decision between "noise present" or "noise absent" in a spatial sound signal. In a continuous mode the noise detection unit is configured to estimate a probability for the noise signal to be present in the spatial sound signal, i.e., a number between 0 and 1 and/or to estimate the noise signal, e.g., by removing voice signal components from the spatial sound signal. The noise detection unit can also be applied to one or more of the sound signals and/or the total space sound signal generated by the environment sound input. The noise detection unit can be arranged downstream to the spatial filterbank, the beamformer, the voice activity detection unit and/or the sound parameter determination unit. Preferably the noise detection unit is arranged downstream to the voice activity detection unit and configured to determine whether a noise signal is present in a respective spatial sound signal. The noise detection unit can be a unit in the electric circuitry or an algorithm performed in the electric circuitry.

In a preferred embodiment the electric circuitry comprises a control unit. The control unit is preferably configured to adaptively adjust subspace parameters (defining a configuration of subspaces), e.g., extension, number, and/or location coordinates, of the subspaces according to the outcome of the voice activity detection unit, sound parameter determination unit and/or the noise detection unit. The adjustment of the extension of the subspaces allows to adjust the form or size of the subspaces. The adjustment of the number of subspaces allows to adjust the sensitivity, respectively resolution and therefore also the computational demands of the hearing system. Adjusting the location coordinates of the subspaces allows to increase the sensitivity at a certain location coordinate or direction in exchange for a decreased sensitivity for other location coordinates or directions. The control unit can for example increase the number of subspaces and decrease the extension of subspaces around a location coordinate of a subspace comprising a voice signal and decrease the number of subspaces and increase the extension of subspaces around a location coordinate of a subspace with a noise signal, with an absence of a sound signal or with a sound signal with a sound level and/or signal-to-noise ratio below a predetermined threshold. This can be favourable for the hearing experience as a user gets a better spatial resolution in a certain direction of interest, while other directions are temporarily of lesser importance. In a preferred embodiment of the hearing system the number of subspaces is kept constant and only the location coordinates and extensions of the subspaces are adjusted, which keeps a computational demand of the hearing system about constant.

In a preferred embodiment the electric circuitry comprises a spatial sound signal selection unit. The spatial sound signal selection unit is preferably configured to select one or more spatial sound signals and to generate an output sound signal from the selected one or more spatial sound signals. The selection of a respective spatial sound signal can for example be based on the presence of a voice signal or noise signal in the respective spatial sound signal, a sound level

and/or a signal-to-noise ratio (SNR) of the respective spatial sound signal. The spatial sound signal selection unit is preferably configured to apply different weights to the one or more spatial sound signals before or after selecting spatial sound signals and to generate an output sound signal from the selected and weighted one or more spatial sound signals. The weighting of the spatial sound signals can be performed on spatial sound signals representing different frequencies and/or spatial sound signals coming from different subspaces, compare also K. L. Bell, et al, "A Bayesian Approach to Robust Adaptive Beamforming," IEEE Trans. Signal Processing, Vol. 4, No. 2, February 2000. Preferably the output transducer is configured to stimulate hearing of a user in dependence of the output sound signal. The spatial sound signal selection unit can be a unit in the electric circuitry or an algorithm performed in the electric circuitry.

In one embodiment the electric circuitry comprises a noise reduction unit. The noise reduction unit is preferably configured to reduce noise in one or more spatial sound signals. Noise reduction for the noise reduction unit is meant as a post-processing step to the noise reduction already performed by spatial filtering and/or beamforming in the spatial filterbanks with beamformers, e.g., by subtracting a noise signal estimated in the noise detection unit. The noise reduction unit can also be configured to reduce noise in the sound signals received by the environment sound input unit and/or the total space sound signal generated from the sound signals. The noise reduction unit can be a unit in the electric circuitry or an algorithm performed in the electric circuitry.

In a preferred embodiment the electric circuitry comprises a user control interface, e.g., a switch, a touch sensitive display, a keyboard, a sensoric unit connected to the user or other control interfaces operable by a user, e.g. fully or partially implemented as an APP of a SmartPhone or similar portable device. The user control interface is preferably configured to allow a user to adjust the subspace parameters of the subspaces. The adjustment of the subspace parameters can be performed manually by the user or the user can select between different modes of operation, e.g., static mode without adaption of the subspace parameters, adaptive mode with adaption of the subspace parameters according to the environment sound received by the environment sound input, i.e., the acoustic environment, or limited-adaptive mode with adaption of the subspace parameters to the acoustic environment which are limited by predetermined limiting parameters or limiting parameters determined by the user. Limiting parameters can for example be parameters that limit a maximal or minimal number of subspaces or the change of the number of subspaces used for the spatial hearing, a maximal or minimal change in extension, minimal or maximal extension, maximal or minimal location coordinates and/or a maximal or minimal change of location coordinates of subspaces. Other modes like modes which fix certain subspaces, e.g., subspaces in front direction and allow other subspaces to be adapted are also possible. In an embodiment, the configuration of subspaces is fixed. In an embodiment, at least one of the subspaces of the configuration of subspaces is fixed. In an embodiment, the configuration of subspaces is dynamically determined. In an embodiment, at least one of the subspaces of the configuration of subspaces is dynamically determined. In an embodiment, the hearing system is configured to provide a configuration of subspaces, wherein at least one subspace is fixed (e.g. located in a direction towards a known target location, e.g. in front of the user), and wherein at least one subspace is adaptively determined (e.g. determined according to the acoustic environment, e.g. in other directions than

iliary devices may be e.g. remote controls, remote microphones, audio gateway devices, mobile phones (e.g. smartphones), public-address systems, car audio systems or music players. Hearing aid devices, hearing aid systems or binaural hearing aid systems may e.g. be used for compensating for a hearing-impaired person's loss of hearing capability, augmenting or protecting a normal-hearing person's hearing capability and/or conveying electronic audio signals to a person.

The hearing aid device may preferably comprise a first wireless interface comprising first antenna and transceiver circuitry adapted for establishing a communication link to an external device and/or to another hearing aid device based on near-field communication (e.g. inductive, e.g. at frequencies below 100 MHz) and/or a second wireless interface comprising second antenna and transceiver circuitry adapted for establishing a second communication link to an external device and/or to another hearing aid device based on far-field communication (radiated fields (RF), e.g. at frequencies above 100 MHz, e.g. around 2.4 or 5.8 GHz).

The invention further resides in a method comprising a step of receiving sound signals representing sound of an environment. Preferably, the method comprises a step of using the sound signals to generate spatial sound signals. Each of the spatial sound signals represents sound coming from a subspace of a total space. The method can alternatively comprise a step of dividing the sound signals in subspaces generating spatial sound signals. A further alternative method comprises a step of generating a total space sound signal from the sound signals and dividing the total space sound signal in subspaces of the total space generating spatial sound signals. The method further preferably comprises a step of detecting whether a voice signal is present in a respective spatial sound signal for all spatial sound signals. The step of detecting whether a voice signal is present in a respective spatial sound signal can be performed one after another for each of the spatial sound signals or is preferably performed in parallel for all spatial sound signals. Preferably, the method comprises a step of selecting spatial sound signals with a voice signal above a predetermined signal-to-noise ratio threshold. The step of selecting spatial sound signals with a voice signal above a predetermined signal-to-noise ratio threshold can be performed one after another for each of the spatial sound signals or is preferably performed in parallel for all spatial sound signals. The spatial sound signals can also be selected based on a sound level threshold or a combination of a sound level threshold and a signal-to-noise ratio threshold. Further in one embodiment spatial sound signals can be selected, which do not comprise a voice signal. The method further preferably comprises a step of generating an output sound signal from the selected spatial sound signals.

A preferred embodiment of the method comprises a step of dividing the sound signals in frequency bands. Dividing the sound signals in frequency bands is preferably performed prior to generating spatial sound signals. The method can comprise a step of reducing noise in the sound signals in the frequency bands and/or noise in the spatial sound signals. Preferably the method comprises a step of reducing noise in the selected spatial sound signals. Preferably the step of reducing noise in the selected spatial sound signals is performed in parallel for all selected spatial sound signals.

In a preferred embodiment the method comprises a step of adjusting subspace parameters of the subspaces. Subspace parameters comprise the extension of the subspace, the number of subspaces and the location coordinates of the subspaces. Preferably the adjusting of the subspace param-

eters of the subspaces is performed in response to the detection of a voice signal or noise signal in a selected spatial sound signal, spatial sound signal or sound signal. The adjusting of the subspace parameters can also be performed manually, e.g., by a user.

A preferred embodiment of the method can be used to determine a location of a sound source. The method preferably comprises a step of receiving sound signals. Preferably the method comprises a step of using the sounds signals and subspace parameters to generate spatial sound signals representing sound coming from a subspace of a total space. The subspaces preferably fill the total space in this embodiment of the method. The method preferably comprises a step of determining a sound level and/or signal-to-noise ratio (SNR) in each spatial sound signal. Preferably, the method comprises a step of adjusting the subspace parameters of the subspaces, which are used for the step of generating the spatial sound signals. The subspace parameters are preferably adjusted such that sensitivity around subspaces with high sound level and/or high signal-to-noise ratio (SNR) is increased and sensitivity around subspaces with low sound level and/or low SNR is decreased. The sensitivity here is to be understood as a resolution of the space, meaning that a higher number of smaller subspaces is arranged in spaces around a sound source, while only a small number of larger subspaces is arranged around or at spaces without a sound source. The method preferably comprises a step of identifying a location of a sound source. The identification of a location of a sound source can depend on a predetermined sound level threshold and/or a predetermined SNR threshold. To reach the predetermined sound level and/or the SNR the method is preferably configured to repeat all steps of the method iteratively until the predetermined sound level and/or the SNR is achieved. The method can also be configured to iteratively adjust the subspace parameters until a change of the subspace parameters is below a threshold value for the change of the sound level and/or the SNR. If the change of the sound level and/or the SNR caused by adjusting the subspace parameters is below a threshold value the location of a sound source is preferably identified as the spatial sound signal with the highest sound level and/or SNR.

In an embodiment, a standard configuration of subspaces is used as an initial configuration. Then sound parameters for all subspaces (spatial sound signals) are determined, e.g., sound level. The subspace with, e.g., highest sound level is the subspace with highest sound source location probability. Then in an iteration step, the subspace with highest sound source location probability is adjusted by, e.g., dividing it in smaller subspaces. The sound level of the smaller subspaces is identified. This is performed until a sound source is located to a degree acceptable for the method or user.

Preferably, the method to determine a location of a sound source comprises a step of determining whether a voice signal is present in the spatial sound signal corresponding to the location of the sound source. If a voice signal is present in the spatial sound signal corresponding to the location of the sound source the method can generate an output sound signal from the spatial sound signal comprising the voice signal and/or spatial sound signals of neighbouring subspaces comprising the voice signal. The output sound signal can be used to stimulate the hearing of a user. Alternatively if no voice signal is present the method preferably comprises a step of identifying another location of a sound source. Preferably the method is performed on a hearing system comprising a memory. After identifying a location of a sound source the method can be manually restarted to identify other sound source locations.

Preferably, the methods described above are performed using the hearing system according to the invention. Further methods can obviously be performed using the features of the hearing system.

The hearing system is preferably configured to be used for sound source localization. The electric circuitry of the hearing system preferably comprises a sound source localization unit. The sound source localization unit is preferably configured to decide if a target sound source is present in a respective subspace. The hearing system preferably comprises a memory configured to store data, e.g., location coordinates of sound sources or subspace parameters, e.g., location coordinates, extension and/or number of subspaces. The memory can also be configured to temporarily store all or a part of the data. The memory is preferably configured to delete the location coordinates of a sound source after a predetermined time, such as 10 seconds, preferably 5 seconds or more preferably 3 seconds.

In a preferred embodiment of the hearing system all detection units are configured to run a hard and a soft mode. The hard mode corresponds to a binary mode, which performs binary decisions between "present" or "not present" for a certain detection event. The soft mode is a continuous mode, which estimates a probability for a certain detection event, i.e., a number between 0 and 1.

The present invention will be more fully understood from the following detailed description of embodiments thereof, taken together with the drawings in which:

FIG. 1 shows a schematic illustration of an embodiment of a hearing system;

FIGS. 2A-2E show schematic illustrations of an embodiment of a hearing system worn by a user listening to sound from a subspace of a total space of the sound environment (FIG. 2A) and four different configurations of subspaces (FIGS. 2B, 2C, 2D, 2E);

FIG. 3 shows a block diagram of an embodiment of a method for processing sound signals representing sound of an environment;

FIG. 1 shows a hearing system 10 comprising a first microphone 12, a second microphone 14, electric circuitry 16, and a speaker 18. The hearing system 10 can also comprise one environment sound input unit that comprises the microphones 12 and 14 or an array of microphones or other sound inputs which are configured to receive incoming sound and generate sound signals from the incoming sound (not shown). Additionally or alternatively to the speaker 18 a cochlear implant can be present in the hearing system 10 or an output transducer configured to stimulate hearing of a user (not shown). The hearing system can also be a binaural hearing system comprising two hearing systems 10 with a total of four microphones (not shown). The hearing system 10 in the embodiment presented in FIG. 1 is a hearing aid, which is configured to stimulate the hearing of a hearing impaired user.

Incoming sound 20 from the environment, e.g., from several sound sources is received by the first microphone 12 and the second microphone 14 of the hearing device 10. The first microphone 12 generates a first sound signal 22 representing the incoming sound 20 at the first microphone 12 and the second microphone 14 generates a second sound signal 24 representing the incoming sound 20 at the second microphone 14. The sound signals 22 and 24 are provided to the electric circuitry 16 via a line 26. In this embodiment the line 26 is a wire that transmits electrical sound signals. The line 26 can also be a pipe, glass fibre or other means for signal transmission, which is configured to transmit data and sound signals, e.g., electrical signals, light signals or other means

for data communication. The electric circuitry 16 processes the sound signals 22 and 24 generating an output sound signal 28. The speaker 18 generates an output sound 30 in dependence of the output sound signal 28.

In the following we describe an exemplary path of processing of the sound signals 22 and 24 in the electric circuitry 16. The electric circuitry 16 comprises a spectral filterbank 32, a sound signal combination unit 33 and a spatial filterbank 34 which comprises several beamformers 36. The electric circuitry 16 further comprises a voice activity detection unit 38, a sound parameter determination unit 40, a noise detection unit 42, a control unit 44, a spatial sound signal selection unit 46, a noise reduction unit 48, a user control interface 50, a sound source localization unit 52, a memory 54, and an output sound processing unit 55. The arrangement of the units in the electric circuitry 16 in FIG. 1 is only exemplary and can be easily optimized by the person skilled in the art for short communication paths if desired.

The processing of the sound signals 22 and 24 in the electric circuitry 16 starts with the spectral filterbanks 32. The spectral filterbanks 32 divide the sound signals 22 and 24 in frequency bands by band-pass filtering copies of the sound signals 22 and 24. The division in frequency bands by band-pass filtering of the respective sound signal 22 and 24 in the respective spectral filterbank 32 can be different in the two spectral filterbanks 32. It is also possible to arrange more spectral filterbanks 32 in the electric circuitry 16, e.g., spectral filterbanks 32 which process sound signals of other sound inputs (not shown). Each of the spectral filterbanks 32 can further comprise rectifiers and/or filters, e.g., low-pass filters or the like (not shown). The sound signals 22 and 24 in the frequency bands can be used to derive spatial information, e.g., by cross correlation calculations. The sound signals 22 and 24 in the frequency bands, i.e., the outputs of the spectral filterbanks 32, are then combined in the sound signal combination unit 33. In this embodiment the sound signal combination unit 33 is configured to generate total subspace sound signals 53 for each frequency band by a linear combination of time-delayed sub-band sound signals, meaning a linear combination of sound signal 22 and sound signal 24 in a respective frequency band. The sound signal combination unit 33 can also be configured to generate a total subspace sound signal 53 or a total subspace sound signal 53 for each frequency band by other methods known in the art to combine the sound signals 22 and 24 in the frequency bands. This allows to perform spatial filtering for each frequency band.

Each total subspace sound signal 53 in a frequency band is then provided to the spatial filterbank 34. The spatial filterbank 34 comprises several beamformers 36. The beamformers 36 are operated in parallel to each other. Each beamformer is configured to use the total subspace sound signal 53 in a respective frequency band to generate a spatial sound signal 56 in a respective frequency band. Each beamformer can also be configured to use a total subspace sound signal 56 summed over all frequency bands to generate a spatial sound signal 56. Each of the spatial sound signals 56 represents sound coming from a subspace 58 of a total space 60 (see FIGS. 2A-2E). The total space 60 is the complete surrounding of a user 62, i.e., the acoustic environment (see FIGS. 2A-2E).

In the following we describe an example situation where the spatial filterbank 34 is especially useful, i.e., a situation in which the sound scene changes, e.g., by occurrence of a new sound source. We here compare our hearing system 10 with a standard hearing aid without a spatial filterbank that

has a single beamformer with a beam pointing in front direction, meaning that the hearing aid mainly receives sound from the front of the head of a user wearing the standard hearing aid. Without the spatial filterbank **34** the user needs to determine the location of the new sound source and adjust the subspace parameters accordingly to receive sound signals. In a sound scene change the beam has to be adjusted from an initial subspace to the subspace of the sound source, meaning that the user wearing the hearing aid has to turn his head from an initial direction to the direction of the new sound source. This takes time and the user risks that he misses, e.g., the onset of the speech of a new talker. With the spatial filterbank **34**, the user already has a beam pointing in the direction or subspace of the sound source; all the user or hearing system **10** needs to do is to decide to feed the respective spatial sound signal **56**, i.e., the respective beamformer output to the user **62**.

The spatial filterbank **34** further allows for soft-decision schemes, where several spatial sound signals **56** from different subspaces **58**, i.e., beamformer outputs from different directions, can be used to generate an output sound signal **28** at the same time. Instead of a hard-decision in terms of listening to one and only one spatial sound signal **56**, it is, e.g., possible to listen to 30% of a spatial sound signal **56** representing a subspace **58** in front of a user, 21% of a second spatial sound signal **56** representing a second subspace **58**, and 49% of a third spatial sound signal **56** representing a third subspace **58**. Such an architecture is useful for systems, where target signal presence in a given subspace or direction is expressed in terms of probabilities. The underlying theory for such a system has been developed in, e.g., K. L. Bell, et al, "A Bayesian Approach to Robust Adaptive Beamforming," IEEE Trans. Signal Processing, Vol. 4, No. 2, February 2000.

There can also be more than one spatial filterbank **34**. The spatial filterbank **34** can also be a spatial filterbank algorithm. The spatial filterbank algorithm can be executed as a spatial filterbank **34** online in the electric circuitry **16** of the hearing system **10**. The spatial filterbank **34** in the embodiment of FIG. **1** uses the Fast Fourier Transform for computing the spatial sound signals **56**, i.e., beams. The spatial filterbank **34** can also use other means, i.e., algorithms for computing the spatial sound signals **56**.

The spatial sound signals **56** generated by the spatial filterbank **34** are provided to the voice activity detection unit **38** for further processing. Each of the spatial sound signals **56** is analysed in the voice activity detection unit **38**. The voice activity detection unit **38** detects whether a voice signal is present in the respective spatial sound signal **56**. The voice detection unit **38** is configured to perform to modes of operation, i.e., detection modes. In a binary mode the voice activity detection unit **38** is configured to make a binary decision between "voice present" or "voice absent" in a spatial sound signal **56**. In a continuous mode the voice activity detection unit **38** is configured to estimate a probability for the voice signal to be present in the spatial sound signal **56**, i.e., a number between 0 and 1. The voice detection is performed according to methods known in the art, e.g., by using a means to detect whether harmonic structure and synchronous energy is present in the respective spatial sound signal **56**, which indicates a voice signal, as vowels have unique characteristics consisting of a fundamental tone and a number of harmonics showing up synchronously in the frequencies above the fundamental tone. The voice activity detection unit **38** can be configured to continuously detect whether a voice signal is present in the respective spatial sound signal **56** or only for selected spatial

sound signals **56**, e.g., spatial sound signals **56** with a sound level above a sound level threshold and/or spatial sound signals **56** with a signal-to-noise ratio (SNR) above a SNR threshold. The voice activity detection unit **38** can be a unit in the electric circuitry **16** or an algorithm performed in the electric circuitry **16**.

Voice activity detection (VAD) algorithms in common systems are typically performed directly on a sound signal, which is most likely noisy. The processing of the sound signals with a spatial filterbank **34** results in spatial sound signals **56** which represent sound coming from a certain subspace **58**. Performing independent VAD algorithms on each of the spatial sound signals **56** allows easier detection of a voice signal in a subspace **58**, as potential noise signals from other subspaces **58** have been rejected by the spatial filterbank **34**. Each of the beamformers **36** of the spatial filterbank **34** improves the target signal-to-noise signal ratio. The parallel processing with several VAD algorithms allows the detection of several voice signals, i.e., talkers, if they are located in different subspaces **58**, meaning that the voice signal is in a different spatial sound signals **56**. The spatial sound signals **56** are then provided to the sound parameter determination unit **40**. The sound parameter determination unit **40** is configured to determine a sound level and/or signal-to-noise ratio of a spatial sound signal and/or if a sound level and/or signal-to-noise ratio of a spatial sound signal **56** is above a predetermined threshold. The sound parameter determination unit **40** can be configured to only determine sound level and/or signal-to-noise ratio for spatial sound signals **56** which comprise a voice signal.

The spatial sound signals **56** can alternatively be provided to the sound parameter determination unit **40** prior to the voice activity detection unit **38**. Then the voice activity detection unit **38** can be configured only to be activated to detect whether a voice signal is present in a spatial sound signal **56** when the sound level and/or signal-to-noise ratio of a spatial sound signal **56** is above a predetermined threshold. The sound parameter determination unit **40** can be a unit in the electric circuitry **16** or an algorithm performed in the electric circuitry **16**.

The spatial sound signals **56** are then provided to the noise detection unit **42**. The noise detection unit **42** is configured to determine whether a noise signal is present in a respective spatial sound signal **56**. The noise detection unit **42** can be a unit in the electric circuitry **16** or an algorithm performed in the electric circuitry **16**.

The spatial sound signals **56** are then provided to the control unit **44**. The control unit **44** is configured to adaptively adjust the subspace parameters, e.g., extension, number, and/or location coordinates of the subspaces according to the outcome of the voice activity detection unit **38**, sound parameter determination unit **40** and/or the noise detection unit **42**. The control unit **44** can for example increase the number of subspaces **58** and decrease the extension of subspaces **58** around a location coordinate of a subspace **58** comprising a voice signal and decrease the number of subspaces **58** and increase the extension of subspaces **58** around a location coordinate of a subspace **58** with a noise signal, with an absence of a sound signal **22** or **24** or with a sound signal **22** or **24** with a sound level and/or signal-to-noise ratio below a predetermined threshold. This can be favourable for the hearing experience as a user gets a better spatial resolution in a certain direction of interest, while other directions are temporarily of lesser importance.

The spatial sound signals **56** are then provided to the spatial sound signal selection unit **46**. The spatial sound signal selection unit **46** is configured to select one or more

spatial sound signals **56** and to generate a weight parameter value for the one or more selected spatial sound signals **56**. The weighting and selection of a respective spatial sound signal **56** can for example be based on the presence of a voice signal or noise signal in the respective spatial sound signal **56**, a sound level and/or a signal-to-noise ratio (SNR) of the respective spatial sound signal **56**. The spatial sound signal selection unit **46** can be a unit in the electric circuitry **16** or an algorithm performed in the electric circuitry **16**.

The spatial sound signals **56** are then provided to the noise reduction unit **48**. The noise reduction unit **48** is configured to reduce the noise in the spatial sound signals **56** selected by the spatial sound signal selection unit **46**. Noise reduction in the noise reduction unit **48** is a post-processing step, e.g., a noise signal is estimated and subtracted from a spatial sound signal **56**. Alternatively all spatial sound signals **56** can be provided to the noise reduction unit **48**, which then reduces the noise in one or more spatial sound signals **56**. The noise reduction unit **48** can be a unit in the electric circuitry **16** or an algorithm performed in the electric circuitry **16**.

The spatial sound signals **56** are finally provided to the output sound processing unit **55** together with all output results, e.g. weight parameters, selection of spatial sound signals **56**, or other outputs determined by the foregoing units in the electric circuitry **16**. The output sound processing unit **55** is configured to process the spatial sound signals **56** according to the output results of the foregoing units in the electric circuitry **16** and generate an output signal **28** in dependence of the output results of the foregoing units in the electric circuitry **16**. The output signal **28** is for example adjusted by, selecting spatial sound signals **56** representing subspaces **58** with voice activity, without feedback, or with/without other properties determined by the units of the electric circuitry **16**. The output sound processing unit **55** is further configured to perform hearing aid processing, such as feedback cancellation, feedback suppression, and hearing loss compensation (amplification, compression) or similar processing.

The output sound signal **28** is provided to the speaker **18** in a final step. The output transducer **18** then generates an output sound **30** in dependence of the output sound signal **28**.

The user **62** can control the hearing system **10** using the user control interface **50**. The user control interface **50** in this embodiment is a switch. The user control interface **50** can also be a touch sensitive display, a keyboard, a sensoric unit connected to the user **62**, e.g., a brain implant or other control interfaces operable by the user **62**. The user control interface **50** is configured to allow the user **62** to adjust the subspace parameters of the subspaces **58**. The user can select between different modes of operation, e.g., static mode without adaption of the subspace parameters, adaptive mode with adaption of the subspace parameters according to the environment sound received by the microphones **12** and **14**, i.e., the acoustic environment, or limited-adaptive mode with adaption of the subspace parameters to the acoustic environment which are limited by predetermined limiting parameters or limiting parameters determined by the user **62**. Limiting parameters can for example be parameters that limit a maximal or minimal number of subspaces **58** or the change of the number of subspaces **58** used for the spatial hearing, a maximal or minimal change in extension, minimal or maximal extension, maximal or minimal location coordinates and/or a maximal or minimal change of location coordinates of subspaces **58**. Other modes like modes which fix certain subspaces **58** and allow other subspaces **58** to be adapted are also possible, e.g., fixing subspaces **58** in front direction and allowing the adaption of all other subspaces **58**. Using an alternative user control interface can allow to adjust the subspace parameters (defining a configuration of subspaces) directly. The hearing system **10** can also be connected to an external device for controlling the hearing system **10** (not shown).

By adaptively adjusting subspace parameters the spatial filterbanks **34** become adaptive spatial filters. The term "adaptive" (in the meaning "adaptive/automatic or user-controlled") is intended to cover two extreme situations: a) signal adaptive/automatic, and b) user-controlled, i.e., the user tells the algorithm in which direction to "listen" and any soft-combination between a) and b), e.g. that the algorithm makes proposals about directions, which the human user accepts/rejects. In an embodiment, a user using the user control interface **50** can select to listen to the output of a single spatial sound signal **56**, which may be adapted to another subspace **58** or subspaces **58**, i.e. directions, than a frontal subspace **58**. The advantage of this is that it allows the listener to select to listen to spatial sound signals **56** which represent sound **20** coming from non-frontal directions, e.g., in a car-cabin situation. A disadvantage in prior art hearing aids is that it takes time for a user, and therefore the beam to change direction, e.g., from frontal, to the side by turning the head of the hearing aid user. During the travelling time of the beam, the first syllable of a sentence may be lost, which leads to reduced intelligibility for a hearing impaired user of the prior art hearing aid. The spatial filterbank **34** covers all subspaces, i.e., directions. The user can manually select or let an automatic system decide, which spatial sound signal **56** or spatial sound signals **56** are used to generate an output sound signal **56**, which is then transformed into an output sound **30**, which can be presented instantly to the hearing aid user **62**.

In one mode of operation the hearing system **10** allows to localize a sound source using the sound source localization unit **52**. The sound source localization unit **52** is configured to decide if a target sound source is present in a respective subspace. This can be achieved using the spatial filterbank and a sound source localization algorithm which zooms in on a certain subspace or direction in space to decide if a target sound source is present in the respective subspace or direction in space. The sound source localization algorithm used in the embodiment of the hearing system **10** presented in FIG. 1 comprises the following steps.

Sound signals **22** and **24** are received.

Spatial sound signals **56** representing sound **20** coming from a subspace **58** of a total space **60** are generated using the sounds signals **22** and/or **24** and subspace parameters. The subspaces **58** in the sound source localization algorithm are chosen to fill the total space **60**. A sound level, signal-to-noise ratio (SNR), and/or target signal presence probability in each spatial sound signal **56** is determined.

The subspace parameters of the subspaces **58**, which are used for the step of generating the spatial sound signals **56** are adjusted. The subspace parameters are preferably adjusted such that sensitivity around subspaces **58** with high sound level and/or high signal-to-noise ratio (SNR) is increased and sensitivity around subspaces **58** with low sound level and/or low SNR is decreased. Also other adjustments of the subspaces **58** are possible.

A location of a sound source is identified. It is also possible that more than one sound source and the locations of the respective sound sources are identified. The identification of a location of a sound source depends on a predetermined sound level threshold and/or a predetermined SNR

threshold. To reach the predetermined sound level and/or the SNR the sound source localization algorithm is configured to repeat all steps of the algorithm, meaning receiving sound signals **22** and **24**, generating spatial sound signals **56**, adjusting subspace parameters and identifying locations of a sound source, iteratively until the predetermined sound level and/or the SNR is achieved. Alternatively the sound source localization algorithm is configured to iteratively adjust the subspace parameters until a change of the subspace parameters is below a threshold value for the change of the sound level and/or the SNR. If the change of the sound level and/or the SNR caused by adjusting the subspace parameters is below a threshold value the location of a sound source is identified as the spatial sound signal **56** with the highest sound level and/or SNR. It is also possible to identify more than one sound source and locations of the respective sound sources in parallel. A further, e.g. second, sound source can be identified as the spatial sound signal **56** with the next, e.g. second, highest sound level and/or SNR. Preferably the spatial sound signals **56** of the sound sources can be compared to each other to identify whether the spatial sound signals come from an identical sound source. In this case the algorithm is configured to process only the strongest spatial sound signal **56**, meaning the spatial sound signal **56** with the highest sound level and/or SNR, representing a respective sound source. Spatial sound signals **56** representing different sound sources can be processed by parallel processes of the algorithm. The total space **60** used for the location of sound sources can be limited to respective subspaces **58** for a respective process of the parallel processes to avoid two sound sources in an identical subspace **58**.

If a sound source is identified the sound source localization algorithm comprises a step of using the respective spatial sound signal **56** representing the sound coming from the subspace **58** of the sound source and optionally spatial sound signals **56** representing sound coming from subspaces **58** which are in close proximity to the subspace **58** of the sound source to generate an output sound signal **28**.

The sound source localization algorithm can also comprise a step of determining whether a voice signal is present in the spatial sound signal **56** corresponding to the location of the sound source.

If a voice signal is present in the spatial sound signal **56** representing the sound coming from the subspace **58** of the sound source the algorithm comprises a step of generating an output sound signal **28** from the spatial sound signal **56** comprising the voice signal and/or spatial sound signals **56** of neighbouring subspaces **58** comprising the voice signal.

Alternatively if no voice signal is present the sound source localization algorithm comprises a step of identifying another location of a sound source. After identifying a location of a sound source the sound source localization algorithm can be manually restarted to identify other sound source locations.

The memory **54** of the hearing system **10** is configured to store data, e.g., location coordinates of sound sources or subspace parameters, e.g., location coordinates, extension and/or number of subspaces **58**. The memory **54** can be configured to temporarily store all or a part of the data. In this embodiment the memory **54** is configured to delete the location coordinates of a sound source after a predetermined time, such as 10 seconds, preferably 5 seconds or more preferably 3 seconds.

Relying on the parallel sound source localization algorithm above, the hearing system **10** can estimate the subspace **58**, i.e. the direction, of a sound source. The direction of a target sound source is of interest, as dedicated noise reduction systems can be applied to enhance signals from this particular direction.

The spatial sound signals **56** generated by the spatial filterbank **34** can also be used for improved feedback howl detection, which is a challenge in any state-of-the-art hearing device. The howling results due to feedback of the loudspeaker signal to the microphone(s) of a hearing aid. The hearing aid has to distinguish between the following two situations: i) a feedback howl, or ii) an external sound signal, e.g., a violin playing, which as a signal looks similar to a feedback howl. The spatial filterbank **34** allows to exploit the fact that feedback howls tend to occur from a particular subspace **58**, i.e. direction, so that the spatial difference between a howl and the violin playing can be exploited for improved howl detection.

The electric circuitry **16** of the hearing system **10** can comprise a transceiver unit **57**. In the embodiment shown in FIG. **1** the electric circuitry **16** does not comprise a transceiver unit **57**. The transceiver unit **57** can be configured to transmit data and sound signals to another hearing system **10**, speakers in another persons hearing aid, in mobile phones, in laptops, in hearing aid accessories, streamers, tv-boxes or other systems comprising a means to receive data and sound signals and receive data and sound signals from another hearing system **10**, an external microphone, external microphones, e.g., microphones in a hearing aid of another user, in mobile phones, in laptops, in hearing aid accessories, audio streamers, audio gateways, tv-boxes e.g. for wirelessly transmitting TV sound, or other systems comprising a means to generate a data and/or sound signal and to transmit data and sound signals. In the case of two hearing systems **10** connected to each other the hearing systems **10** form a binaural hearing system. All filterbanks and/or units, meaning **32**, **34**, **36**, **40**, **42**, **44**, **46**, **48**, **50**, **52**, and/or **54** of the electric circuitry **16** can be configured for binaural usage. All of the units can be improved by combining the output of the units binaurally. The spatial filterbanks **34** of the two hearing systems can be extended to binaural filter banks or the spatial filterbanks **34** can be used as binaural filterbanks, i.e., instead of using 2 local microphones **12** and **14**, the binaural filter banks are configured to use four sound signals of four microphones. The binaural usage improves the spectral and spatial sensitivity, i.e., resolution of the hearing system **10**. A potential transmission time delay between the transceiver units **57** of the two hearing systems **10**, which can typically be between 1 to 15 ms depending on the transmitted data, is of no practical concern, as the sound source localization units **52** are used for sound source localization or voice activity detection units **38** are used for detection purpose in the case of binaural usage of the hearing system. The spatial sound signals **56** are then selected in dependence of the output of the respective units. The decisions of the units can be delayed 15 ms without any noticeable performance degradations. In another embodiment the output sound signal is generated from the output of the units. The units, filterbanks and/or beamformers can also be algorithms performed on the electric circuitry **16** or a processor of the electric circuitry **16** (not shown).

FIG. **2A** shows the hearing system of FIG. **1** worn by a user **62**. The total space **60** in this embodiment is a cylinder volume, but may alternatively have any other form. The total space **60** can also for example be represented by a sphere (or semi-sphere, a dodecahedron, a cube, or similar geometric structures. A subspace **56** of the total space **60** corresponds to a cylinder sector. The subspaces **58** can also be spheres,

cylinders, pyramids, dodecahedra or other geometrical structures that allow to divide the total space **60** into subspaces **58**. The subspaces **58** in this embodiment add up to the total space **60**, meaning that the subspaces **58** fill the total space **60** completely and do not overlap (as e.g. schematically illustrated in FIG. **2**B, each $beam_p$, p=1, 2, . . . , P, constituting a subspace (cross-section) where P (here equal to 8) is the number of subspaces **58**). There can also be empty spaces between subspaces **56** and/or overlap of subspaces **56**. The subspaces **56** in this embodiment are equally spaced, e.g., in 8 cylinder sectors with 45 degrees. The subspaces can also be differently spaced, e.g., one sector with 100 degree, a second sector with 50 degree and a third sector with 75 degree. In one embodiment the spatial filter-bank **34** can be configured to divide the sound signals **22** and **24** in subspaces **56** corresponding to directions of a horizontal "pie", which can be divided into, e.g., 18 slices of 20 degrees with a total space **60** of 360 degrees. In this embodiment the output sound **30** presented to the user **62** by the speaker **18** is generated from an output sound signal **28** that comprises the spatial sound signal **56** representing the subspace **58** of the total space **60**. The subspaces may (in particular modes of operation) be either fixed, or dynamically determined, or a mixture thereof (e.g. some fixed, other adaptively determined).

The location coordinates, extension, and number of subspaces **58** depends on subspace parameters. The subspace parameters can be adaptively adjusted, e.g., in dependence of an outcome of the voice activity detection unit **38**, the sound parameter determination unit **40** and/or the noise detection unit **42**. The adjustment of the extension of the subspaces **58** allows to adjust the form or size of the subspaces **58**. The adjustment of the number of subspaces **58** allows to adjust the sensitivity, respectively resolution and therefore also the computational demands of the hearing system **10**. Adjusting the location coordinates of the subspaces **58** allows to increase the sensitivity at certain location coordinates or direction in exchange for a decreased sensitivity for other location coordinates or directions. In the embodiment of the hearing system **10** in FIGS. **2**A-**2**E the number of subspaces **58** is kept constant and only the location coordinates and extensions of the subspaces are adjusted, which keeps a computational demand of the hearing system about constant.

FIGS. **2**C and **2**D illustrate application scenarios comprising different configurations of subspaces. In FIG. **2**C, the space **60** around the user **62** is divided into 4 subspaces **58**, denoted $beam_1$, $beam_2$, $beam_3$, $beam_4$ in FIG. **2**C. Each subspace beam comprises one fourth of the total angular space, i.e. each spanning 90° (in the plane shown), and each being of equal form and size. The subspaces need not be of equal form and size, but can in principle be of any form and size (and location relative to the user). Likewise, the subspaces need not add up to fill the total space, but may be focused on continuous or discrete volumes of the total space. In FIG. **2**D, the subspace configuration comprises only a part of the space around the user **62** (here a fourth, here subspace $beam_4$ in FIG. **2**C is divided into 2 subspaces **58**, denoted $beam_{41}$, $beam_{42}$ in FIG. **2**D).

FIGS. **2**C and **2**D may illustrate a scenario where the acoustic field in a space around a user is analysed in at least two steps using different configurations of the subspaces of the spatial filterbank, e.g. first and second configurations, and where the second configuration is derived from an analysis of the sound field in the first configuration of subspaces, e.g. according to a predefined criterion, e.g. regarding characteristics of the spatial sound signals of the

configuration of subspaces. A sound source S is shown located in a direction represented by vector d_s relative to the user **62**. The spatial sound signals ($sssig_i$, i=1, 2, 3, 4) of the subspaces **58** of a given configuration of subspaces (e.g. $beam_1$, $beam_2$, $beam_3$, $beam_4$ in FIG. **2**C) is e.g. analysed to evaluate characteristics of each corresponding spatial sound signal (here no prior knowledge of the location and nature of the sound source S is assumed). Based on the analysis, a subsequent configuration of subspaces is determined (e.g. $beam_{41}$, $beam_{42}$ in FIG. **2**D), and the spatial sound signals ($sssig_{ij}$, i=4, j=1, 2) of the subspaces **58** of the subsequent configuration are again analysed to evaluate characteristics of each (subsequent) spatial sound signal. In an embodiment, characteristics of the spatial sound signals comprise a measure comprising signal and noise (e.g. a signal to noise signal to noise ratio). In an embodiment, characteristics of the spatial sound signals comprise a measure representative of a voice activity detection. In an embodiment, a noise level is determined in time segments where no voice is detected by the voice activity detector. In an embodiment, a signal to noise ratio (S/N) is determined for each of the spatial sound signals ($sssig_i$, i=1, 2, 3, 4). The signal to noise ratio (S/N($sssig_4$)) of subspace $beam_4$ is the largest of the four S/N-values of FIG. **2**C, because the sound source is located in that subspace (or in a direction from the user within that subspace). Based thereon, the subspace of the first configuration (of FIG. **2**C) that fulfills the predefined criterion (subspace for which $sssig_i$, i=1, 2, 3, 4 has MAX(S/N)) is selected and further subdivided into a second configuration of subspaces aiming at possibly finding a subspace, for which the corresponding spatial sound signal has an even larger signal to noise ratio (e.g. found by applying the same criterion that was applied to the first configuration of subspaces). Thereby, the subspace defined by $beam_{42}$ is identified as the subspace having the largest signal to noise ratio. An approximate direction to the source is automatically defined (within the spatial angle defined by subspace $beam_{42}$). If necessary a third subspace configuration based on $beam_{42}$ (or alternatively or additionally a finer subdivision of the subspaces of configuration **2** (e.g. more than two subspaces)) can be defined and the criterion for selection applied.

In the above example, the predefined criterion for selecting a subspace or the corresponding spatial sound signal was maximum signal to noise ratio. Other criteria may be defined, e.g. minimum signal to noise ratio or a predefined signal to noise ratio (e.g. in a predefined range). Other criteria may e.g. be based on maximum probability for voice detection, or minimum noise level, or maximum noise level, etc.

FIG. **2**E illustrates a situation where the configuration of subspaces comprises fixed as well as adaptively determined subspaces. In the example shown in FIG. **2**E a fixed subspace ($beam_{1F}$) is located in a direction d_s towards a known target sound source S (e.g. a person or a loudspeaker) in front of the user **62**, and wherein the rest of the subspaces (cross-hatched subspaces $beam_{1D}$ to $beam_{6D}$) are adaptively determined, e.g. determined according to the current acoustic environment. Other configurations of subspaces comprising a mixture of fixed and dynamically (e.g. adaptively) determined subspaces are possible.

FIG. **3** shows an embodiment of a method for processing sound signals **22** and **24** representing incoming sound **20** of an environment. The method comprises the following steps.

100 Receiving sound signals **22** and **24** representing sound **20** of an environment.

110 Using the sound signals **22** and **24** to generate spatial sound signals **56**. Each spatial sound signal **56** represents sound **20** coming from a subspace **58** of a total space **60**.

120 Detecting whether a voice signal is present in a respective spatial sound signal **56** for all spatial sound signals **56**. The step **120** is preferably performed in parallel for all spatial sound signals **56**.

130 Selecting spatial sound signals **56** with a voice signal above a predetermined signal-to-noise ratio threshold. The step **130** is performed in parallel for all spatial sound signals **56**.

140 Generating an output sound signal **28** from the selected spatial sound signals **56**.

Alternatively, the step **110** can be dividing the sound signals in subspaces **58** generating spatial sound signals **56**. A further alternative for step **110** is generating a total space sound signal from the sound signals **56** and dividing the total space sound signal in subspaces **58** of the total space **60** generating spatial sound signals **56**.

The step **120** of detecting whether a voice signal is present in a respective spatial sound signal **56** can also be performed one after another for each of the spatial sound signals **56**.

The step **130** of selecting spatial sound signals with a voice signal above a predetermined signal-to-noise ratio threshold can also be performed one after another for each of the spatial sound signals **56**. The spatial sound signals **56** can also be selected based on a sound level threshold or a combination of a sound level threshold and a signal-to-noise ratio threshold. Further in an alternative embodiment spatial sound signals **56** can be selected, which do not comprise a voice signal.

REFERENCE SIGNS

10 hearing system
12 first microphone
14 second microphone
16 electric circuitry
18 speaker
20 incoming sound from the environment
22 first sound signal
24 second sound signal
26 line
28 output sound signal
30 output sound
32 spectral filterbank
33 sound signal combination unit
34 spatial filterbank
36 beamformer
38 voice activity detection unit
40 sound parameter determination unit
42 noise detection unit
44 control unit
46 spatial sound signal selection unit
48 noise reduction unit
50 user control interface
52 sound source localization unit
54 memory
55 output sound processing unit
56 spatial sound signals
57 transceiver unit
58 subspace
60 total space
62 user

The invention claimed is:

1. A hearing system configured to be worn by a user, the hearing system comprising:

an environment sound input unit;

an output transducer; and

electric circuitry,

wherein the environment sound input unit is configured to receive sound from the environment of the environment sound input unit and to generate sound signals representing sound of the environment,

wherein the output transducer is configured to stimulate hearing of the user,

wherein the electric circuitry comprises a spatial filterbank configured to use the sound signals to generate spatial sound signals dividing a total space of the environment sound in a plurality of subspaces, defining a configuration of the subspaces,

wherein a spatial sound signal represents sound coming from a subspace, and

wherein the hearing system is configured to provide a configuration of the subspaces wherein at least one subspace is fixed and wherein at least one subspace is adaptively determined.

2. A hearing system according to claim **1**, wherein the spatial filterbank comprises at least one beamformer configured to process the sound signals by generating a spatial sound signal which represents sound coming from a subspace.

3. A hearing system according to claim **1**, wherein the subspaces are cylinder sectors or cones of a sphere.

4. A hearing system according to claim **1**, wherein the subspaces add up to the total space.

5. A hearing system according to claim **1**, wherein the subspaces of the plurality of subspaces are equally spaced.

6. A hearing system according to claim **1**, wherein the electric circuitry comprises a spatial sound signal selection unit configured to select one or more spatial sound signals and generate an output sound signal from the selected one or more spatial sound signals, and wherein the output transducer is configured to stimulate hearing of a user in dependence of the output sound signal.

7. A hearing system according to claim **6**, wherein the spatial sound signal selection unit is configured to weight the selected one or more spatial sound signals and generate an output sound signal from the selected and weighted one or more spatial sound signals.

8. A hearing system according to claim **1**, wherein the electric circuitry comprises a noise reduction unit configured to reduce noise in one or more spatial sound signals.

9. A hearing system according to claim **1**, wherein the electric circuitry comprises at least one spectral filterbank configured to divide the sound signals in frequency bands.

10. A hearing system according to claim **1** comprising a hearing aid configured to stimulate the hearing of a hearing impaired user.

11. A hearing system configured to be worn by a user, the hearing system comprising:

an environment sound input unit;

an output transducer; and

electric circuitry,

wherein the environment sound input unit is configured to receive sound from the environment of the environment sound input unit and to generate sound signals representing sound of the environment,

wherein the output transducer is configured to stimulate hearing of the user,

wherein the electric circuitry comprises a spatial filter-bank configured to use the sound signals to generate spatial sound signals dividing a total space of the environment sound in a plurality of subspaces, defining a configuration of the subspaces,

wherein a spatial sound signal represents sound coming from a subspace, and

wherein the electric circuitry further comprises a control unit configured to dynamically adjust the configuration of the subspaces.

12. A hearing system according to claim 11, wherein the electric circuitry comprises a voice activity detection unit configured to determine whether a voice signal is present in a respective spatial sound signal, and/or a noise detection unit configured to determine whether a noise signal is present in, or to determine a level of noise of, a respective spatial sound signal.

13. A hearing system according to claim 11 configured to provide a configuration of the subspaces wherein at least one subspace is fixed and wherein at least one subspace is adaptively determined.

14. A hearing system configured to be worn by a user, the hearing system comprising:
an environment sound input unit;
an output transducer; and
electric circuitry,
wherein the environment sound input unit is configured to receive sound from the environment of the environment sound input unit and to generate sound signals representing sound of the environment,
wherein the output transducer is configured to stimulate hearing of the user,
wherein the electric circuitry comprises a spatial filter-bank configured to use the sound signals to generate spatial sound signals dividing a total space of the environment sound in a plurality of subspaces, defining a configuration of the subspaces,
wherein a spatial sound signal represents sound coming from a subspace,
wherein the electric circuitry further comprises a voice activity detection unit configured to determine whether a voice signal is present in a respective spatial sound signal, and/or a noise detection unit configured to determine whether a noise signal is present in, or to determine a level of noise of, a respective spatial sound signal, and
wherein the electric circuitry further comprises a control unit configured to adaptively adjust the configuration of the subspaces according to the output of the voice activity detection unit and/or the noise detection unit.

15. A hearing system configured to be worn by a user, the hearing system comprising:
an environment sound input unit;
an output transducer; and
electric circuitry,
wherein the environment sound input unit is configured to receive sound from the environment of the environment sound input unit and to generate sound signals representing sound of the environment,

wherein the output transducer is configured to stimulate hearing of the user,

wherein the electric circuitry comprises a spatial filter-bank configured to use the sound signals to generate spatial sound signals dividing a total space of the environment sound in a plurality of subspaces, defining a configuration of the subspaces,

wherein a spatial sound signal represents sound coming from a subspace, and

wherein the electric circuitry further comprises a user control interface configured to allow a user to adjust the configuration of the subspaces.

16. A hearing system configured to be worn by a user, the hearing system comprising:
an environment sound input unit;
an output transducer; and
electric circuitry,
wherein the environment sound input unit is configured to receive sound from the environment of the environment sound input unit and to generate sound signals representing sound of the environment,
wherein the output transducer is configured to stimulate hearing of the user,
wherein the electric circuitry comprises a spatial filter-bank configured to use the sound signals to generate spatial sound signals dividing a total space of the environment sound in a plurality of subspaces, defining a configuration of the subspaces,
wherein a spatial sound signal represents sound coming from a subspace, and
wherein the hearing system is configured to analyse the sound signals representing sound of the environment in at least a first and a second step using first and second different configurations of subspaces by the spatial filterbank in the first and second steps, respectively, and where the second configuration is derived from an analysis of spatial sound signals of the first configuration of the subspaces.

17. A method for processing sound signals representing sound of an environment with a hearing system, comprising:
receiving sound signals representing sound of an environment with an environment sound input unit of the hearing system;
using the sound signals with a spatial filterbank of the hearing system to generate spatial sound signals dividing a total space of the environment sound in a plurality of subspaces, wherein each spatial sound signal represents sound coming from a subspace of a total space;
defining a configuration of the subspaces;
detecting with a voice activity detection unit whether a voice signal is present in a respective spatial sound signal for all spatial sound signals;
adaptively adjusting the configuration of the subspaces according to the output of the voice activity detection unit and/or a noise detection unit;
selecting spatial sound signals with a voice signal above a predetermined signal-to-noise ratio threshold; and
generating an output sound signal from the selected spatial sound signals.

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